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Commissioners of the
Federal Communications Commission
c/o William J. Tricarico, Secretary
Room 222
1919 M Street, N.W.
Washington, DC 20554

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Ref: MM Docket No. 87-268

Gentlemen:

Our company wishes to submit the enclosed comments regarding the FCC inquiry into Advanced Television Systems and Their Impact on the Existing Television Broadcast Service. We applaud the Commission on the manner in which it is approaching this matter, and anticipate an outcome of major benefit to the U.S. public.

Sincerely,

Brit Conner

Brit Conner
President & CEO

encl.

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Comments on MM Docket No. 87-268

In the Matter of:

Advanced Television Systems and Their Impact on the Existing Television Broadcast Service

Review of Technical and Operational Requirements: Part 73-E, Television Broadcast Stations

Reevaluation of the UHF Television Channel and Distance Separation Requirements of Part 73 of the Commission's Rules

Digideck, Incorporated supports and encourages the Commission in its inquiry into future uses of the spectrum by advanced television systems.

Summary

Digideck is a small, start-up company engaged in research on data compression techniques for digital audio. Part of our studies have included serious research into the feasibility of compatible digital sound transmission for the broadcast television industry. Our research indicates that several techniques offer the potential for improved sound quality over the existing BTSC sound system, while maintaining compatibility with the BTSC signal, and using receiver decoding techniques which are feasible with current hardware capabilities.

Further, our inquiries among major video producers have indicated the market desirability of some form of multi-channel sound (surround-sound), a multi-lingual capability, and a near "CD-like" sound quality to go with any future enhanced video. For these reasons we encourage the Commission to seriously consider the inclusion of an improved audio system coincident with the specification of an improved video format.

If our research is correct, and it is now feasible to upgrade the existing audio in a compatible manner, then it makes more sense to make that upgrade now, coincident with the specification of an improved video format. Otherwise one risks the potential obsolesence of the "improved receivers" when the added features become available through an augmentation channel.

Specific Comments on Compatible Digital Audio (in reference to Docket Questions 1-4)

Improvements to the BTSC audio system would appear to be easily implementable in an augmentation channel of either 3- or 6-MHz. Some pushing and shoving to achieve the right fit with respect to the augmentation video will be inevitable, but most researchers appear to agree that an improved audio system is easily achieved by use of the additional spectrum.

The challenge is to find a method of improving the existing sound system within the existing spectrum, without replacing the analog MTS signal.

This challenge translates into three primary issues:

(1) How large a digital data rate can the existing spectrum support compatibly (and what portion of this can be devoted to audio)?

(2) Can a digital audio signal maintain an "improved fidelity" while being restricted to this data rate?

(3) Is the signal processing associated with this technique feasible and reasonable within the manufacturing cost standards of the near future?

Studies within the last few years have indicated the feasibility of compatibly supporting some 750- to 850-Kbps in a digital data stream within the NTSC format. Proposed techniques include addition of a new audio subcarier (1), use of a subcarrier in quadrature with the video carrier (2), and insertion of data bits into the vertical and/or horizontal blanking intervals (3). Verification of the compatibility under field conditions remains an issue, and some of the proposed approaches may be in conflict with proposed signal schemes for improved video. Nevertheless there appears to be a general consensus that there is potential for inserting digital data at these rates.

The BTSC system achieves a 14-KHz stereo SNR in the range of 60-64 db, absent certain transmission and receiver anomalies, such as multipath and filter tolerances or nonlinearities (4), roughly equivalent to 10- to 12-bit digital audio. Thus a target for an improved audio system could be set at the equivalent of 14-bit audio, though many may prefer to pursue 16-bit audio quality.

If the 14-KHz bandwidth is considered to be an adequate target, and three channels are considered to be a requirement (to at least offer a match to the MTS design), then the above data rate would imply a restriction to about 250 Kbps per channel, or about 7.9 bits per sample (31.5 KHz sample rate) including all error correction, overhead and ancillary data. A tradeoff, based on field trials, must be made between the bits allocated to signal quality versus those allocated to error protection; nevertheless it would appear that an allocation of no more than 20 to 25 percent of the overall stream would be needed for error protection, leaving some 6.0 to 6.3 bits per sample for the compressed signal itself.

Starting from experience in voice compression systems for the telephone industry, researchers in digital audio have investigated and proposed systems covering companding (2), adaptive delta modulation (5), adaptive differential PCM (ADPCM) (6), and various transform and filter bank techniques (7),(8). Each of these systems has been shown to be tantalizingly close to the above goals, but to be marginal in some aspect:

Companding to six bits achieves, at best, a signal-to-noise ratio of perhaps seven to eight bits. The primary advantage of companding is its simplicity, low cost and instantaneous computation, but the overall sound quality is lacking.

Adaptive delta modulation can generally meet the above objectives, provided the audio content falls within the generally encountered spectrum of "low frequencies dominant". The primary advantages of ADM are its relative simplicity, low receiver cost and relatively short processing delay. The sound quality (at the equivalent of six bits per sample) is right on the borderline between generally adequate and marginal, falling short on certain infrequent passages, such as hand-bells. At 4 or 5 bits per sample the performance of ADM is totally inadequate.

ADPCM at a rate of six bits per sample has an actual SNR equivalent to six bits (approximately 36 dB), but because the noise is buried beneath the signal in a spectrally matched manner, the perceived noise level is improved somewhat over "white" noise. Implementation of ADPCM is a bit of a computational challenge at the receiver by today's hardware, requiring perhaps four to eight floating point multiplies per data point. While clearly superior to companding, ADPCM still lacks the SNR to perform as a true "high fidelity" signal.

Transform coding and other filter bank techniques, usually coupled with psychoacoustic design rules, have, until recently proved highest in resulting signal quality. Reports of excellent sound quality at rates as low as four bits per sample are in the literature. The principal disadvantage of transform techniques is the requirement for an inverse transform at the receiver, a computational complexity far beyond present day hardware limits.

Because of this near ability to meet the design goals, many have been tempted to stretch these techniques to their limit. A better approach is now possible.

A <u>heretofore undisclosed development</u> combines entropy coding with adaptive noiseless compression filters, which are then used in conjunction with noise shaped truncation to fit to the allowable channel rate. This new approach, developed originally for use in recording precision medical waveforms by researchers at SRI International in the mid- to late-1970's, has been considerably upgraded by our company for use with digital audio and was slated for announcement to the industry in mid-1988. Concurrent with these comments Digideck is preparing a limited public unveiling to support the Commission's inquiry.

When run at six bits per sample, this new entropy code has an average SNR, using C-weighting, on the order of 85-90 dB, well within or above the target range. Further, even under worst case conditions, it does not alter the amplitude and phase of the input signal, unlike ADM, and thus maintains proper interchannel relationships for items such as stereo separation and surround sound matrix decoders. Finally, the implementation is computationally simple, unlike the transform and filter bank concepts. No floating point operations are needed, and in the worst case only eight shifts and sixteen adds per data point are required by the receiver. Laboratory computer based simulations of this algorithm have been running for several months at Digideck, and certain major television equipment manufacturers have completed real-time hardware implementations of a preliminary nature. The feasibility of custom chip implementation for a television receiver has been proven, and refinement efforts to lower production costs are underway.

Digideck will be prepared to work with interested industry participants in support of research to address the issues raised by the Commission's inquiry.

References:

(1) Craig C. Todd, "Digital Sound and Data for Broadcast Television -- A Compatible System", 1987 NAB Engineering Conference Proceedings.

(2) "Digital Audio Signal Multiplex to NTSC with full compatibility", Consumer Products Research Center, Hitachi, Ltd.

(3) Private communications with a major manufacturer. Also suggested in (2).

(4) J. James Gibson, "Effects of Receiver Design and Transmission Impairments on Audio Signal Quality in the BTSC System for Multichannel Television Sound", Proc. of AES 4th International Conference (Rosemont, Ill, May 1986)

(5) Craig C. Todd, "Efficient Digital Audio Coding & Transmission Systems", paper adapted from talk given at 75th Convention of the Audio Engineering Society (Paris, March 1984).

(6) Private communications with another major manufacturer.

(7) E. F. Schroeder and H.-J. Platte,"'MSC': Stereo Audio Coding with CD-Quality and 256 Kbit/sec", paper presented at 1987 ICCE, Chicago.

(8) G. Theile, M. Link and G. Stoll, "Low-bit rate coding of high quality audio signals", paper presented at 82nd Convention of the Audio Engineering Society (London, March 1987).